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**Digital Enhanced Cordless Telecommunications (DECT);
Study of Super Wideband Codec in DECT for narrowband,
wideband and super-wideband audio communication
including options of low delay
audio connections (≤ 10 ms framing)**

Reference

DTR/DECT-00316

Keywords

codec, DECT, superwideband

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Digital Enhanced Cordless Telecommunications (DECT).

Modal verbs terminology

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Introduction

Since the introduction of additional codecs in New Generation DECT [i.5] in 2007, wideband services have been widely established for fixed line, mobile and OTT communications networks. This trend is gaining even more momentum by services using cutting edge codecs like 3GPP EVS and the upcoming new Bluetooth™ codec by offering super-wideband audio bandwidth.

NOTE: Bluetooth™ is the trade name of a wireless technology standard for exchanging data over short distances (using short-wavelength UHF radio waves in the ISM band from 2,4 - 2,485 GHz) from fixed and mobile devices, and building personal area networks (PANs), owned by the Bluetooth Special Interest Group. This information is given for the convenience of users of the present document and does not constitute an endorsement by ETSI of the technology named. Equivalent technologies may be used if they can be shown to lead to the same results.

Recent market research from several relevant DECT infrastructure providers indicates a demand for upgrading DECT services and standard with additional features enabled by evolved speech and audio codecs.

The present document collects performance requirements to add a real benefit to current and upcoming DECT applications and evaluates the Low Complexity Communication Codec (LC3) on suitability for this as well as discusses possible adaptations for DECT environments in terms of error protection and signalling.

1 Scope

The present document provides a study of technical updates to the DECT standard to enable super wideband (SWB) audio calls in existing DECT slot formats as well as technical improvements to narrowband (NB) and wideband (WB) calls. All required change requests are listed and defined for the different DECT layers to enable high quality audio communication between DECT FP and PP including DECT repeaters (relays). The study includes an investigation on FEC for block-based codecs. Information is provided on the audio quality in some DECT use cases for NB, WB and SWB and potential improvements by a new audio codec are studied.

2 References

2.1 Normative references

Normative references are not applicable in the present document.

2.2 Informative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long-term validity.

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] ETSI EN 300 175-1: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 1: Overview".

[i.2] ETSI EN 300 175-8: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech and audio coding and transmission".

[i.3] EP2901594B1: "Error Detection for sub-band ADPCM encoded sound signal".

[i.4] ETSI EN 300 175-3: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 3: Medium Access Control (MAC) layer".

[i.5] ETSI TS 102 527-3: "Digital Enhanced Cordless Telecommunications (DECT); New Generation DECT; Part 3: Extended wideband speech services".

[i.6] ETSI EN 300 700: "Digital Enhanced Cordless Telecommunications (DECT); Wireless Relay Station (WRS)".

[i.7] ETSI EN 300 175-5: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 5: Network (NWK) layer".

[i.8] ETSI EN 300 176-2: "Digital Enhanced Cordless Telecommunications (DECT); Test specification; Part 2: Audio and speech".

[i.9] Recommendation ITU-T P.863 (09-2014): "Perceptual objective listening quality assessment".

[i.10] Recommendation ITU-T P.800 (08-1996): "Methods for subjective determination of transmission quality".

[i.11] Recommendation ITU-T G.191 (03-1996): "Software tools for speech and audio coding standardization" .

[i.12] Recommendation ITU-T G.722 (09-2012): "7 kHz audio-coding within 64 kbit/s".

[i.13] Recommendation ITU-T G.726 (12-1990): "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)".

[i.14] ETSI TS 126 071: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Mandatory speech CODEC speech processing functions; AMR speech Codec; General description (3GPP TS 26.071)".

[i.15] ETSI TS 126 171: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; General description (3GPP TS 26.171)".

[i.16] ETSI TS 126 441: "Universal Mobile Telecommunications System (UMTS); LTE; Codec for Enhanced Voice Services (EVS); General overview (3GPP TS 26.441)".

[i.17] SIG Bluetooth™ Hands-Free Profile 1.7.1.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in ETSI EN 300 175-1 [i.1] and the following apply:

Fullband (FB): speech or audio sampled at 48 kHz

Fullband, compact disc (FBCD): speech or audio sampled at 44,1 kHz

Narrowband (NB): speech or audio sampled at 8 kHz

Semi-Super Wideband (SSWB): speech or audio sampled at 24 kHz

Super Wideband (SWB): speech or audio sampled at 32 kHz

Wideband (WB): speech or audio sampled at 16 kHz

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

3GPP 3rd Generation Partnership Project
ACR Absolute Category Rating

NOTE: See Recommendation ITU-T P.800 [i.10].

AMR Adaptive Multi-Rate,

NOTE: See ETSI TS 126 071 [i.14].

AMR-WB Adaptive Multi-Rate Wideband

NOTE: See ETSI TS 126 171 [i.15].

BCH Bose-Chaudhuri-Hocquenghem
BER Bit Error Rate
CELT Constrained Energy Lapped Transform
CMR Codec Mode Request
CNG Comfort Noise Generation
CRC Cyclic Redundancy Check
CS Circuit Switched
DECT Digital Enhanced Cordless Telecommunications

DSP Digital Signal Processor
DTX Discontinuous Transmission
EP Error Protection
ETSI European Telecommunications Standards Institute
EVS Enhanced Voice Service

NOTE: See ETSI TS 126 441 [i.16].

FB Fullband
FBCD Fullband Compact Disc
FEC Forward Error Correction
FER Frame Erasure Rate
FP Fixed Part (DECT bas station)
FT Frame Type
GFSK Gaussian Frequency-Shift Keying
GSM Global System for Mobile Communications
IETF Internet Engineering Task Force
I_{NA} higher layer Information channel (unprotected), minimum delay operation

NOTE: See ETSI TS 102 527-3 [i.5].

I_{NB} higher layer Information channel (unprotected), normal delay operation

NOTE: See ETSI TS 102 527-3 [i.5].

IP Internet Protocol
LAN Local Area Network
LC3 Low Complexity Communication Codec
MIPS Million Instructions Per Second
MNRU Modulated Noise Reference Unit
MOS-LQO Mean Opinion Score - Listening Quality Objective
mSBC modified Subband Coding

NOTE: See [i.17].

NB Narrowband
OTT Over-The-Top content
PLC Packet Loss Concealment
PLR Packet Loss Rate
PP Portable Part (DECT handset)
RAM Random-Access Memory
RSSI Received Signal Strength Indicator
RTP Real-Time Protocol
SSWB Semi-Super Wideband
STL Software Tool Library
SWB Super Wideband
VoIP Voice over Internet Protocol
WB Wideband
WMOPS Weighted Millions of Operations Per Second
WRS Wireless Relay Station

4 Investigations on an enhanced DECT codec

4.1 Overview

The investigations are organized as follows:

1) Definition of general required codec features (clause 4.2).

2) Study on error profiles and protection schemes for DECT systems (clause 4.3).

3) Characterization of the LC3 as potential candidate (clause 4.4).

4) Definition of required update to DECT specifications (clause 4.5).

4.2 Design constraints/features to be supported

4.2.1 Improved exploitation on DECT slots

Table 1 compares the slot related requirements of the legacy DECT codecs with the proposed new DECT codec.

Table 1: Overview of DECT slot related requirements for new codec

Slot usage	Legacy DECT codecs	Enhanced DECT codec
Normal slots	NB calls (G.726)	NB and WB calls
Long slots	WB calls (G.722)	WB and SWB calls

NB and WB audio quality should be comparable to or better relative to the legacy DECT codecs.

The user experience for error prone channels is expected to be comparable to or better relative to the legacy DECT codecs.

4.2.2 Transmission latency

The codec should operate on 10 ms frame sizes as provided by the DECT transmission slots [i.1]. On top of the framing delay of 10 ms, the additional algorithmic delay should be less than or equal to 2,5 ms.

Additionally, the codec should support frame sizes of 5 ms as well to enable new low delay application besides telephony. For instance, in-room conferencing/amplification or parliament systems require a microphone to loudspeaker delay of less than 20 ms. This guarantees lip-synchronism of the speaker to the amplified signal.

4.2.3 Supported sampling rates, audio bandwidths and sample depths

The codec should support NB, WB, SWB and FB audio bandwidths at the native sample rates of 8 kHz, 16 kHz, 32 kHz and 48 kHz. Additionally, 24 kHz (SSWB) should be supported.

The codec should support the coding of lower audio bandwidths for a given sample rate, e.g. coding of NB signals at 32 kHz. The codec should support the coding of audio samples with 16 bits per samples and may support coding of audio samples with 24 bits per sample.

4.2.4 Support for music streaming

The codec should provide decent audio quality for music streaming services and may provide additional coding features to support stereo music channels.

4.2.5 Packet loss concealment

The codec should support packet loss concealment without adding further algorithmic delay. As the main application is voice, the packet loss concealment should perform well for speech signals.

4.2.6 Low codec complexity

The codec should run with a low computational complexity and low memory footprint to be implementable on typical DECT handheld devices. The complexity should be measured and reported using the latest ITU-T STL complexity measurement toolbox [i.11].

4.3 Investigations on impact of block-based codec

4.3.1 Probability and distribution of bit errors

4.3.1.1 Normal slots - transmission error profile I

An error profile was measured using a real DECT system simulating that a DECT caller is moving through an office building. The caller starts close to the base station and walks away through the office. The measurement recorded the number of bit errors, the position of the bit error, the signal strength (RSSI, in steps of eight) and complete frame losses. Figure 1 outlines the error profile where complete frame losses are indicated by 384 bit errors (A+B field) in combination with signal strength zero.

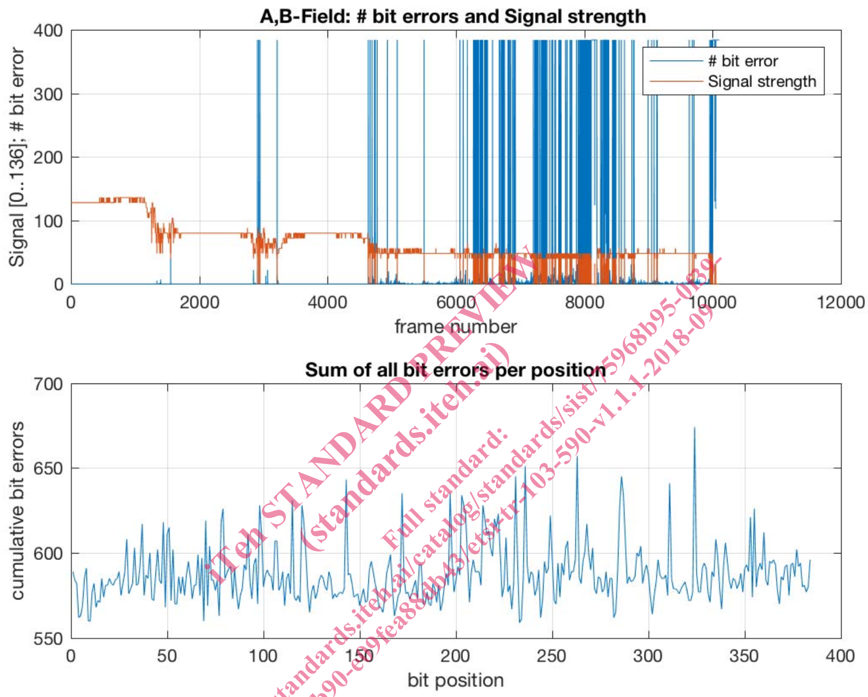

Figure 1: Characterization of Error Profile I

Regarding the position of the bit errors inside the frame, no specific dependency can be found. In order to structure the analysis of the pattern, the bit error profile is further grouped into certain signal strength classes as outlined in Figure 2.

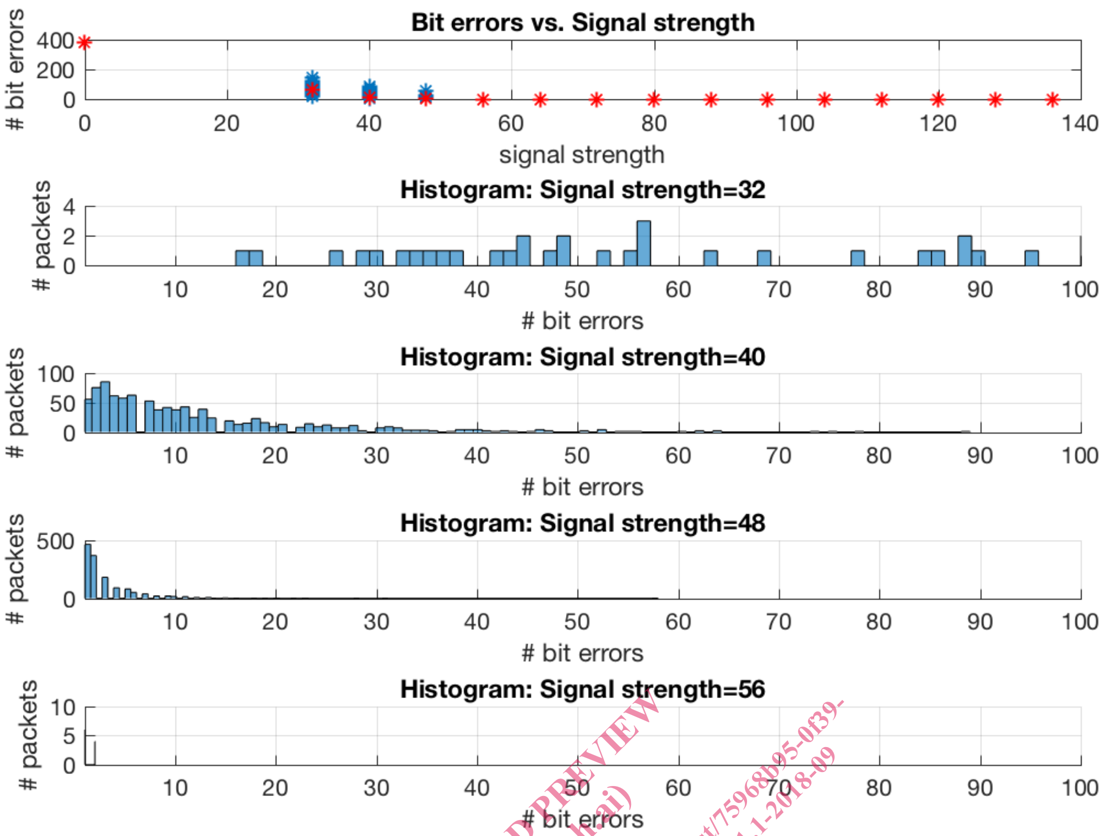

Figure 2: Plot of absolute number (blue) and average number (red) of bit errors for each signal strength (top); Histogram of number of bit errors inside frame for signal strength levels 32 dB, 40 dB, 48 dB and 56 dB

The plots show that:

- For this profile, bit errors only occur for a signal strength level ≤ 56 dB.
- For the signal strength level 48 dB and 56 dB, most packets show less than three bit errors in one packet.
- For signal strength level 40 dB, the bit error rate per packet is significantly higher compared to level 48 dB.
- For signal strength level 32 dB, almost all bits are affected; this level is thus not considered for any recovery activity.
- Four different error protection classes may be appropriate to address the different error characteristics depending on the signal strength, i.e. clean channel, 56 dB, 48 dB, 40 dB.

The packet loss rate (PLR) per signal strength can be estimated by averaging the signal strength over one second. Figure 3 shows packet losses in relation to the averaged signal strength.

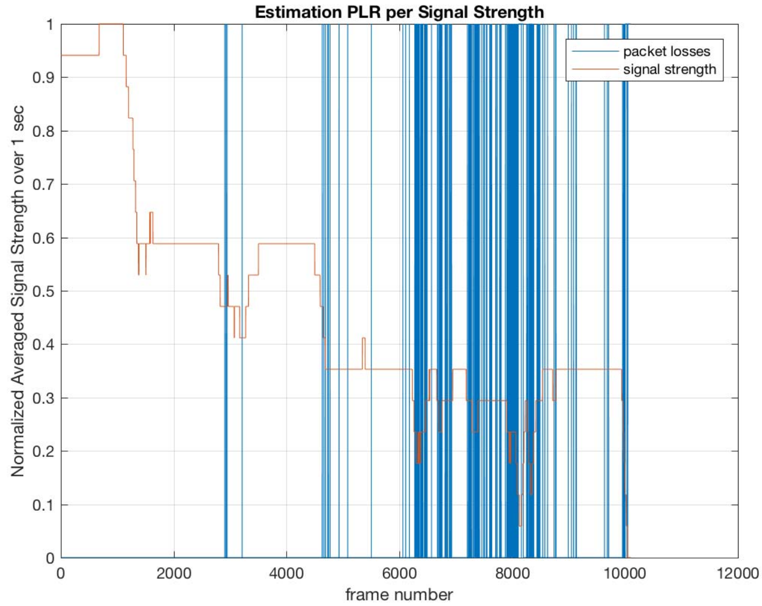

Figure 3: Packet loss rate estimation

According to the given data, a specific PLR can be assigned to a certain signal level as outlined in Table 2.

Table 2: Packet loss and bit error rates

Normalized Averaged signal strength (RSSI)	# Frames	# Packet losses	PLR [%]	PLR rounded [%]	BER rounded [%]
1 (136 dB)	430	0	0	0	
0,94 (128 dB)	728	0	0	0	
0,88 (120 dB)	41	0	0	0	
0,82 (112 dB)	78	0	0	0	
0,76 (104 dB)	20	0	0	0	
0,71 (96 dB)	28	0	0	0	
0,65 (88 dB)	75	0	0	0	
0,59 (80 dB)	2 380	0	0	0	
0,53 (72 dB)	320	0	0	0	
0,47 (64 dB)	436	5	1,15	1	
0,41 (56 dB)	203	2	0,99	1	0,01
0,35 (48 dB)	3 195	28	0,88	1	0,31
0,29 (40 dB)	1 190	88	7,39	7	2,92
0,24 (32 dB)	543	133	24,49	24	17,13

Please note that for realistic simulations packet losses come in addition to any bit errors. Figure 4 shows the cumulative bit error probability as number of bit errors per frame for the relevant RSSIs. This plot provides a compact view of the expected number of bit errors per packet, e.g. at RSSI 48, about 62 % of the packets show no bit error, 70 % show less than 2 errors and 80 % of the packets show less than 4 bit errors.